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Patentanmeldung Nr. Patent application No. Demande de brevet n°

02024084.2

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If no title is shown please refer to the description.
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RTCP reporting for multi-user services in wireless networks

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1. TITLE

RTCP Reporting for Multi-User Services in Wireless Networks

Technical field of the invention

The present invention relates to a method for performing multicast in a telecommunication network.

Especially is the present application applicable in a point-to-point packet-switched telecommunication network.

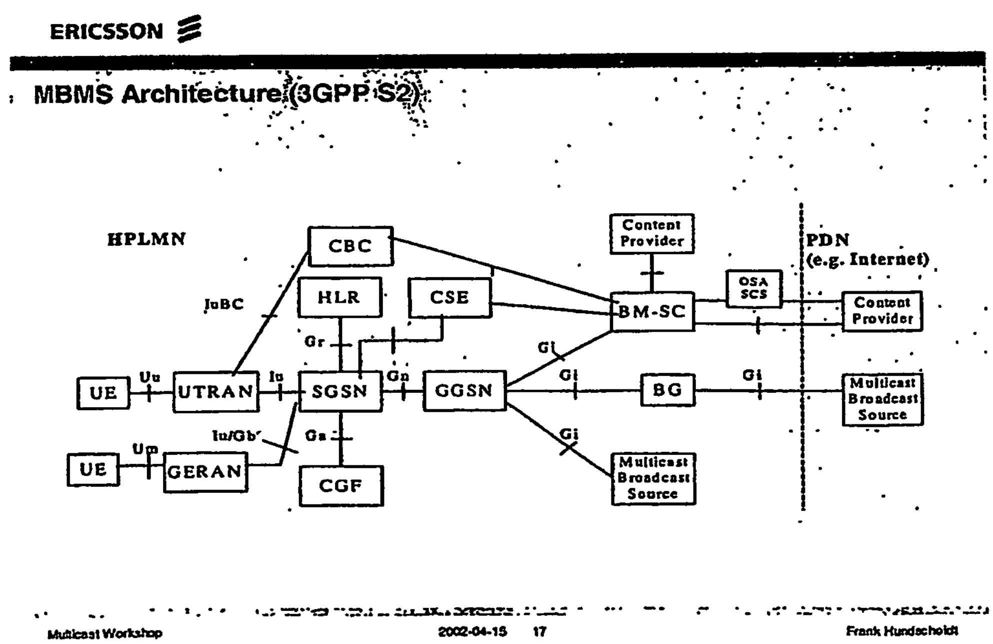
2. BACKGROUND

Universal Mobile Telecommunication System UMTS is being developed to offer wireless wideband multimedia service using Internet protocol. The UMTS as a third-generation 3G mobile communication combines streaming with a range of unique services, like for example geographical positioning, to provide high-quality Internet content to the users. Images, voice, audio and video content are example of mobile multimedia services, which are delivered to the users via media streaming and download techniques. It means once the content has been put onto a media server, it can be delivered on-demand via download or streaming. To download content, the user clicks on a link and waits for the content to be downloaded and playback to begin. Download capabilities are easy to integrate since the hypertext transfer protocol (HTTP) can be used for downloading files. To access streaming

data, the user clicks on a link to start playback, which is almost immediate. Because streaming is a semi-real time service that receives and plays back data at the same time, it puts greater demands on protocols and service implementation, especially when the service is to work over networks with little or no quality of service, like this is the case in UMTS. The radio resource, which are used on the last part of a transmission is to be used in a better way.

2.1 MULTIMEDIA BROADCAST/MULTICAST SERVICE (MBMS)

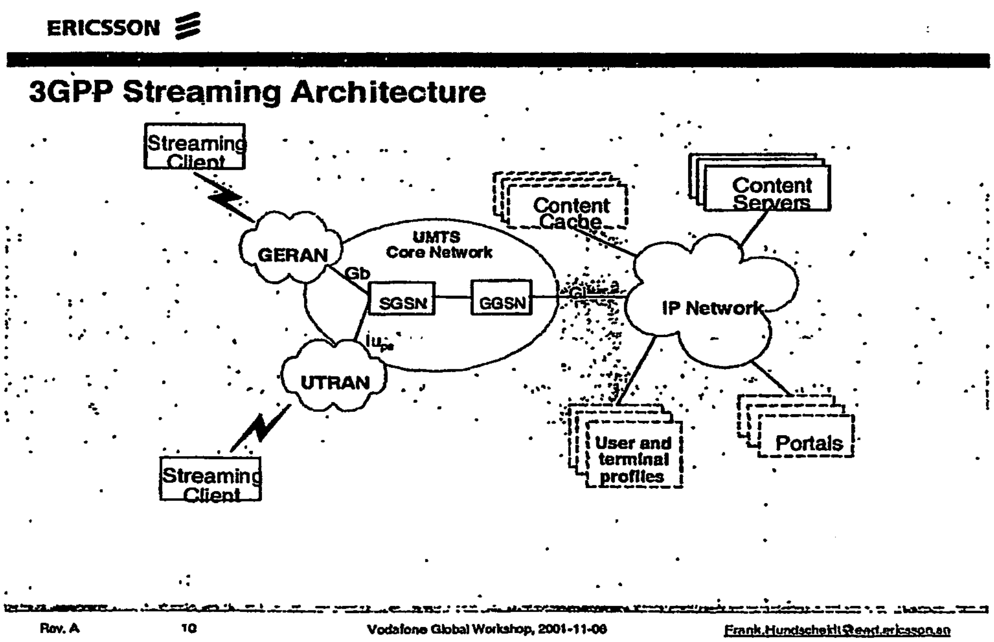
MBMS is a 3GPP release 6 work item, which introduces broadcasting and multicasting into WCDMA and GSM wireless networks. Both broadcast and multicast provide transport efficiency and reduce the load on the content servers (e.g. streaming servers).
The following figure shows the current MBMS architecture, introducing a new network entity, called the Multicast Broadcast Service Center (MB-SC).



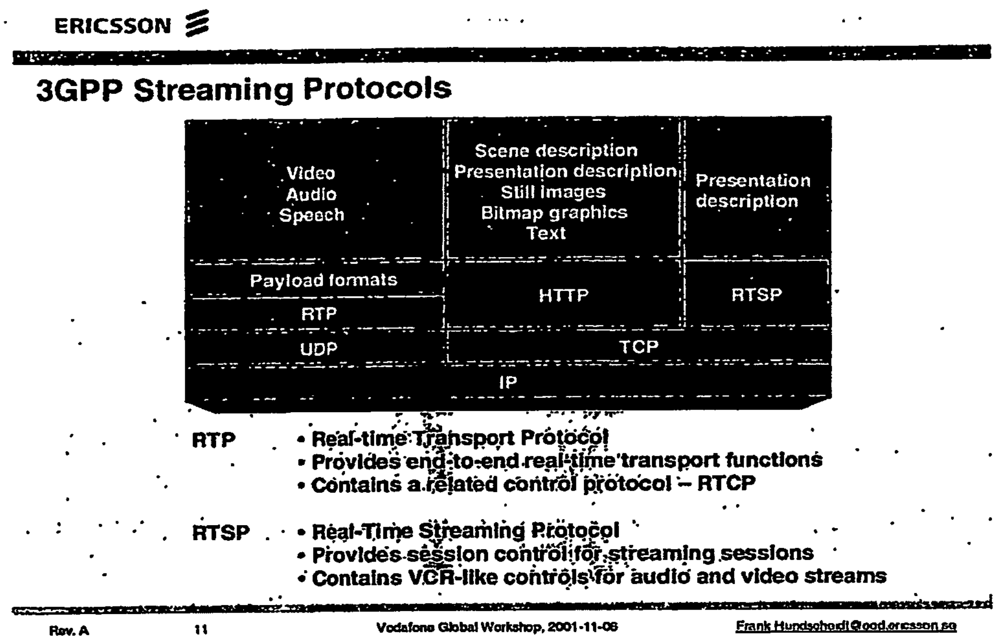
The current intention is to standardize a 'simple' broadcast solution in the radio network. Such a solution implies that there is only a downlink channel for data traffic (i.e. a common/shared downlink channel) but no uplink channel. Thus, no RTCP reports can be sent from the UEs (group members). Note that this radio network solution is not fully settled and that additional RTCP reports or limited RTCP reports (see below for more details) may still apply.

2.2 PACKET SWITCHED STREAMING SERVICE (PSS)

PSS specifies the 3GPP architecture, network entities and protocols for streaming services in wireless networks [1], [2]. The 3GPP PSS architecture is reflected below.



The following figure shows the streaming protocols [2]. RTP is used to transport the dynamic payload, such as video, audio and data.



2.3 REAL-TIME TRANSPORT (CONTROL) PROTOCOL (RTP/RTCP)

The Real-time Transport Protocol (RTP) provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. The functions provided by RTP include payload type identification, sequence numbering, timestamping, and delivery monitoring. RTP is also discussed in the Audio/Video Transport working group in IETF, as *the* protocol for real-time transmission of audio and video over UDP and IP multicast.

The data transport is augmented by a control protocol (RTCP), which is used to monitor the QoS and to convey information about the participants in an ongoing session. Each media stream in a conference is transmitted as a separate RTP session (with a separate RTCP stream). RTCP reports provide statistics about the data received from a particular source,

such as the number of packets lost since the previous report, the cumulative number of packets lost, the interarrival jitter, etc. An additional draft defines a format for extensions to the RTCP sender and receiver reports.

RTP Control Protocol - RTCP (Introduction to Chapter 6 of RFC1889)

The RTP control protocol (RTCP) is based on the periodic transmission of control packets to all participants in the session, using the same distribution mechanism as the data packets. The underlying protocol must provide multiplexing of the data and control packets, for example using separate port numbers with UDP. RTCP performs four functions:

1. The primary function is to provide feedback on the quality of the data distribution. This is an integral part of the RTP's role as a transport protocol and is related to the flow and congestion control functions of other transport protocols. The feedback may be directly useful for control of adaptive encodings [8,9], but experiments with IP multicasting have shown that it is also critical to get feedback from the receivers to diagnose faults in the distribution. Sending reception feedback reports to all participants allows one who is observing problems to evaluate whether those problems are local or global. With a distribution mechanism like IP multicast, it is also possible for an entity such as a network service provider who is not otherwise involved in the session to receive the feedback information and act as a third-party monitor to diagnose network problems. This feedback function is performed by the RTCP sender and receiver reports, described below in Section 6.3.

2. RTCP carries a persistent transport-level identifier for an RTP source called the canonical name or CNAME, Section

6.4.1. Since the SSRC identifier may change if a conflict is discovered or a program is restarted, receivers require the CNAME to keep track of each participant. Receivers also require the CNAME to associate multiple data streams from a given participant in a set of related RTP sessions, for example to synchronize audio and video.

3. The first two functions require that all participants send RTCP packets, therefore the rate must be controlled in order for RTP to scale up to a large number of participants. By having each participant send its control packets to all the others, each can independently observe the number of participants. This number is used to calculate the rate at which the packets are sent, as explained in Section 6.2.

4. A fourth, optional function is to convey minimal session control information, for example participant identification to be displayed in the user interface. This is most likely to be useful in "loosely controlled" sessions where participants enter and leave without membership control or parameter negotiation. RTCP serves as a convenient channel to reach all the participants, but it is not necessarily expected to support all the control communication requirements of an application. A higher-level session control protocol, which is beyond the scope of this document, may be needed.

Functions 1-3 are mandatory when RTP is used in the IP multicast environment, and are recommended for all environments. RTP application designers are advised to avoid mechanisms that can only work in unicast mode and will not scale to larger numbers.

RTCP Transmission Interval (Chapter 6.2 of RFC1889)
[...]

For each session, it is assumed that the data traffic is subject to an aggregate limit called the "session bandwidth" to be divided among the participants. This bandwidth might be reserved and the limit enforced by the network, or it might just be a reasonable share. The session bandwidth may be chosen based or some cost or a priori knowledge of the available network bandwidth for the session. It is somewhat independent of the media encoding, but the encoding choice may be limited by the session bandwidth. The session bandwidth parameter is expected to be supplied by a session management application when it invokes a media application, but media applications may also set a default based on the single-sender data bandwidth for the encoding selected for the session. The application may also enforce bandwidth limits based on multicast scope rules or other criteria.

[...]

The control traffic should be limited to a small and known fraction of the session bandwidth: small so that the primary function of the transport protocol to carry data is not impaired; known so that the control traffic can be included in the bandwidth specification given to a resource reservation protocol, and so that each participant can independently calculate its share. It is suggested that the fraction of the session bandwidth allocated to RTCP be fixed at 5%. While the value of this and other constants in the interval calculation is not critical, all participants in the session must use the same values so the same interval will be calculated. Therefore, these constants should be fixed for a particular profile.

[...]

Maintaining the number of session members

Calculation of the RTCP packet interval depends upon an estimate of the number of sites participating in the session.

New sites are added to the count when they are heard, and an entry for each is created in a table indexed by the SSRC or CSRC identifier (see Section 8.2 of rfc1889) to keep track of them. New entries may not be considered valid until multiple packets carrying the new SSRC have been received (see Appendix A.1 of rfc1889). Entries may be deleted from the table when an RTCP BYE packet with the corresponding SSRC identifier is received.

Both RTP and RTCP have been engineered for multicast.

Definitions from [RFC1889]

Mixer: An intermediate system that receives RTP packets from one or more sources, possibly changes the data format, combines the packets in some manner and then forwards a new RTP packet. Since the timing among multiple input sources will not generally be synchronized, the mixer will make timing adjustments among the streams and generate its own timing for the combined stream. Thus, all data packets originating from a mixer will be identified as having the mixer as their synchronization source.
Translator: An intermediate system that forwards RTP packets with their synchronization source identifier intact. Examples of translators include devices that convert encodings without mixing, replicators from multicast to unicast, and application-level filters in firewalls.

Currently a new, revised version of the RTP/RTCP specification is under preparation in the IETF working group mmusic [5]. Main change compared to the current version (only those, which are relevant for this invention) is, that RTCP receiver reporting can be suppressed. RTCP sender reporting is still mandatory. It is necessary because of inter-media

synchronization (e.g. lip-synchronization when audio and video are transmitted separately) and for sender identification. Suppressing of RTCP packet is necessary for Unidirectional Link Support like in case of digital Broadcast systems.

More specific details about processing of RTCP packets in Translators and Mixers is in [RFC1889] chapter 7.
Problem of the Existing Technology

3. PROBLEM OF THE EXISTING TECHNOLOGY

RTP/RTCP was mainly designed to enable IP based video and audio conferencing services. All participants connected via fixed, bi-directional connections. In conferencing scenarios, all members would like to know, who is actually part of the groups. In particular senders would like to know, who is listening. Anonymous groups with one sender and several thousands of receivers were not really in the scope.

In mobile scenarios usually one-to-many data distribution scenarios are of major interest. One sender in the fixed part on the network sends information to a very large group of users. The group size varies between several thousands and millions of mobile receivers. According to the RTCP specification, each member of the group knows the entire set group members (all other receivers), who also just would like to receive the information. Knowing all group members is important for session control but also to calculate the RTCP report rate. Knowing all other participants of a multicast session is important for conferencing scenarios, but unwanted in one-to-many information distribution scenarios. It is very likely, that receivers would like to stay anonymous. In case

the receiver pays for incoming traffic volume, receiver information is even more unwanted.

The RTCP transmission interval to send reports to the sender and also to other receivers is derived from the group size of the session. The RTCP transmission interval shall be increased with increasing size of the multicast group, i.e. with increasing number of receivers. The dependency prevents the sender from being bombed by too many RTCP messages. With a fixed RTCP transmission rate, the number of RTCP-packets in the system would increase linearly with increasing number of group members and overload each sender. Therefore, each participant needs to receive RTCP packets form neighboring participants.

The one-to-many data distribution scenario is also depicted in the following figure. Note, Receiver 2 and Receiver 3 send their RTCP packet (RR for RTCP Receiver Report) like Receiver 1 to the entire group. All reports from receivers (according to rfc 1889 "Receiver Reports") contain in this scenario one reception report block, since only one sender is active. The sender on the other side sends so-called "sender reports". In this scenario, the sender report contains zero reception report block. This sender is the only sender.

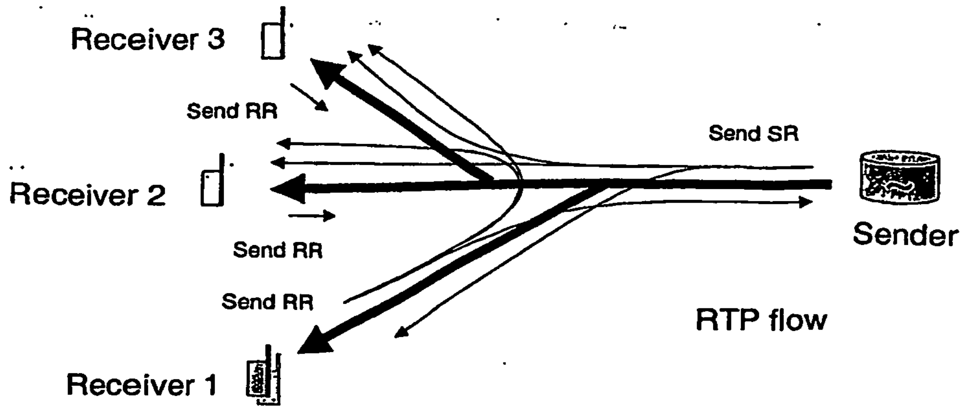


Figure 1: One-Sender, three receiver scenario

In case of radio networks with scarce resources and dense client populations (i.e. many group members in the same geographical area such as a cell), the multicast reporting will waste air interface resources and potentially overloads the network servers (in case of suppressing RTCP packets to the entire group to save radio resources). At the beginning of a multicast session there is also a risk of temporary RTCP overload. Since the group members need reports from other users to adjust their reporting intervals, it may take some time before this saturates in a system with relatively high latency links (compared to fixed networks). This depends also on the initial value for RTCP reporting interval.

Furthermore, in the multicast solution as preferred by Ericsson (and others) there will only be a downlink and no uplink channel. This implies that the members cannot send RTCP reports to the source. It is anyhow questionable what the source should do if an RTCP report from a single client indicates that several RTP frames were corrupted or lost and that client would better be served with a lower data rate. This is especially a big question mark in wireless networks with increasingly heterogeneous clients. In current multicast

solutions the source will either ignore such reports or adapt to the 'slowest' receiver. Both are not adequate when clients are charged for the service and the bearer that is used for the service (as in WCDMA and GSM networks).

So, basically there are the following problems regarding the RTCP reporting in multicast sessions in wireless networks:

- Providing efficient quality feedback to the sender
- Sending receiver information (Name etc) only to the sender instead of the entire group. As soon as the other receiver don't get group size information, the RTCP transmission interval is calculated wrong
- Inefficient use of scarce radio resources (but also backbone resources)
- Risk of server (RNC, SGSN, GGSN, etc.) overload (e.g. 10000 members in a football arena)
- Source must adapt to slowest client (problem in case of heterogeneous networks)
- Long delay before the source is aware of problems (like for the single-user service case)
- RTCP reports contain only info relevant for single client. This implies that the source needs to perform a lot of reporting analyses in case of large user sessions. This is especially a waste if the members can be grouped into one or a few access networks with very similar characteristics and behavior
- RTCP reports don't contain all necessary information for wireless networks

4. SOLUTION

4.1 BASIC IDEA

This invention is mainly about how to generate RTCP receiver reports (from scratch) on an intermediate node based on information from the radio network and/or the client. The intermediate node can either be an RNC or a node at the Gi interface, which receives information from the RNC.

RTCP receiver reports are either generates only based on information from the RNC or based on additional information from the client application.
RTCP can be used in a modified way between the client application and the intermediate node to relay information additional information. RTCP messages are sent on event-basis. The additional information from the client application, which are received at the intermediate node, can impact the radio resource management (e.g. increase FACH power)
By changing the standard RTCP handling in multicast cases in the way of this invention, anonymity between users is archived.

The invention overcomes the problems as listed in chapter 3 by having the first server or any intermediate node in the wireless network taking care of an 'aggregated' RTCP reporting, rather than that each individual takes care of its own reports. RTCP receiver reports contain important feedback about reception conditions from the client. RTCP receiver reports are additionally important for congestion control mechanisms. Therefore, the sender of the RTP multicast stream shall still receive RTCP receiver reports.

The basic idea is to have an intermediate node in the wireless network taking care of an 'aggregated' RTCP reporting, rather than that each user individual performs its own reports. The aggregated' RTCP reporting or general one feedback report per multicast group is generated. A real time

determination of distribution characteristics is performed considering cell related characteristics determination and the group structure and. The generation of the feedback report is based on the real time determination of distribution characteristics wherein said feedback report includes additional information including the number of users to which said feedback report applies. Said feedback report is sent to the multicast source, which utilizes the group feedback report by considering the percentage of the users for which said feedback report applies. The multicast/broadcast transmission is adapted accordingly to the utilized feedback report.

Summary:
The present invention provides mechanism for efficient and intelligent service feedback generation (e.g. RTCP receiver reports), reporting and utilization for multicast and broadcast sessions, which conveys the following:

1.Determination of group structures. E.g. a low speed and a high speed multicast group. (optional)
2.Negotiation of "feedback reporting mechanism" between the network and the clients. (optional)
3.Real-time determination of distribution characteristics
•Cell related characteristics determination
•On the level of a geographical area (one or several radio cells) <u>and</u>
•applicable to a group of clients (e.g. all clients in the same cell, all clients in the same group of cells AND with similar conditions, such as same UE type, same service QoS, same time of registration)
4.Generation of one feedback report per group (at least not one per client), with additional info, such as:
•The number of clients to which the report applies

•Group-related adaptation proposal, considering the characteristics of the access network (e.g. WCDMA, WLAN). E.g. only propose multiple unicast if that's possible, propose handover overlay cell or other access network, etc. This statement means to have the RTCP generating node (e.g. RNC)make a proposal to the source on how to adapt the stream for the corresponding group of users. The reason for this is that the RNC may have information (e.g. radio characteristics of the cell area) that the source does not have and it may thus be in a better situation to judge how the stream could/should be adapted. As an example, RTCP currently indicates that the reception quality is not good, which triggers a source to go down with the bitrate (to the next lower level). An RNC that determines an almost overloaded cell (and many session initiations per time unit) could indicate that the bitrate has to be decreased by 50%, rather than going to the next lower lower (e.g. 80% of the current bitrate).

5.Utilization of the group feedback report by the source (or a proxy), considering the percentage of clients for which the feedback applies, to

•Announce a new "channel" to the clients (e.g. unicast or other access network)

•Adapt the stream (e.g. reduce bit-rate, switch to more reliable codec)

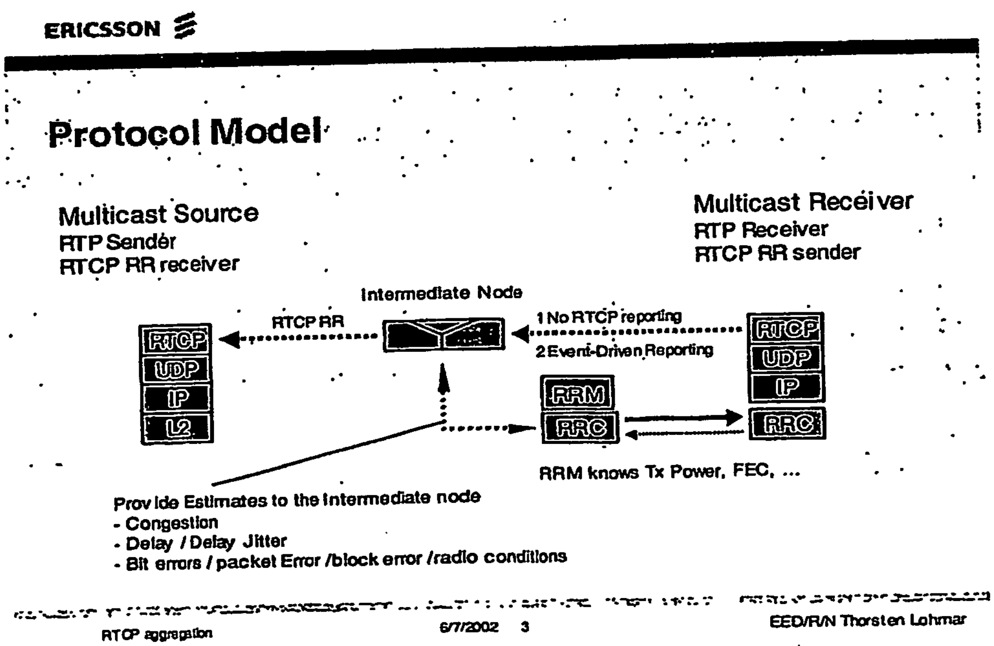
Furthermore, optionally the RTCP report could contain (in addition to the number of users)the actual end-user addresses, since this may be of interest to the other users and/or the source.

The following picture summarizes the three main options of the invention:

1. No RTCP reporting over the air interface and generation of RTCP messages from scratch in an Intermediate node. Reports are generated on the basis of the received RTP stream and potentially on RRM knowledge about a certain cell or area.

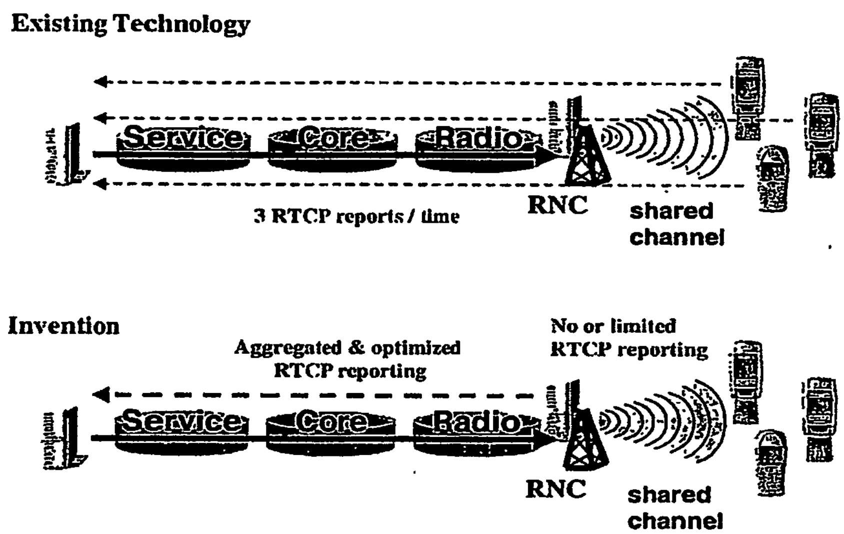
2. In certain exceptional situations the UE is allowed to send RTCP reports on unicast uplink channels. These are then used as additional input to form an aggregated RTCP message as in alternative 1.

3. Another option of the invention is to refrain from sending RTCP reports to all multicast receivers in order to maintain anonymity between the users. In addition this reduces the downlink load in the network. Typically, users are also not interested in who is receiving the information as well. Thus, RTCP reports except the Sender Reports are not sent in downlink direction.



The RNC compiles a single RTCP report from the information that is available from the lower layer protocols and mechanisms and pretends to be a single client. Note that the existing RTCP reporting as defined in [1] may be enhanced to reflect aggregated RTCP reports.

The basic idea is reflected in the following figure.



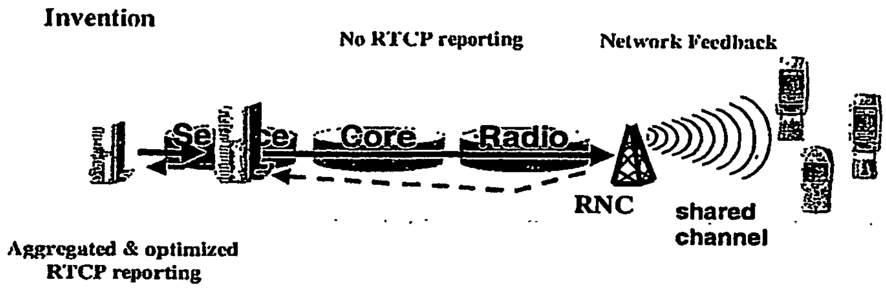
For the reporting from the clients itself, this invention discusses 4 alternatives, each optimized for a different multicast service scenario.

1. The client refrains from sending RTCP reports on RTP/RTCP level. This is currently discussed as a proposal for the standards. This is obviously not new nor inventive.
2. All RTCP reports are blocked in the UE (no impacts on RTCP level)

3. For 1 and 2 there is an option to send only RTCP reports with special information. I.e. no regular RTCP reports are sent. Only in case of extraordinary events, the RTCP reporting is done. In 1. this is handled on RTCP level, whereas in 2. the UE would have to filter the regular RTCP reports. Which RTCP reports are regarded as extraordinary is service and access network dependent and may be subject for negotiation between the source and the destinations and/or between the access networks and the destinations.

4. To refrain the clients from sending RTCP Receiver Reports and generate RTCP receiver reports in an intermediate, enhanced node. This node basically acts like a mixer and hides the receivers of an entire cell. The extended mixer gets information from the RNC and creates a RTCP report from this. Additionally, RTCP packets with special wireless information can be used and filtered out at the RTCP generation node (see 3).

The following figure depicts an alternative, logical architecture. The functionality of getting the necessary quality information at cell level is physically split from the node, which is compiling and sending the RTCP reports. No RTCP receiver reports (sender reports are still mandatory) are sent from the mobile client. Special node (or function in e.g. GGSN; MRF, etc) outside the mobile core generates the RTCP receiver reports. RTCP receiver reports are sent as per-RNC or per-cell aggregate. This allows the sender to collect and evaluate the quality feedback (maybe adapt (sub-)streams), but is omits any session participations information to other mobile clients.



By providing per-cell or per-RNC RTCP reports also increases the periodicity of quality feedback information and allows a sender to adapt faster to changing conditions. The RTCP reporting interval depends on the number of participants in the session.

4.2 TECHNICAL DESCRIPTION

There exist different detailed solutions for different scenarios.

4.2.1 Broadcast Scenario

The broadcast scenario is characterized by a radio access set-up where no up-link is present. This means that the multicast datagrams are sent in downlink direction, but no client response is sent back.
In the current MBMS discussion this is seen as the first step to support multicast in WCDMA.

The lack of a return channel prohibits that RTCP feedback is sent back from the clients. Even though - depending on how protocols stacks are built - RTCP messages might be generated there exist no medium to deliver these.

However, for the current RTP RFC (RFC1889), the used of RTCP receiver reports is mandatory in multicast sessions. In particular it is important that these are received by the multicast sender to signal that clients are still listing. The current revision of the RTP rfc (draft-ietf-avt-rtp-new-11.txt) [6] provides a feature to suppress RTCP receiver report usage. However, by omitting RTCP receiver reports also an important means of getting quality feedback from the receivers is omitted. The source cannot adapt to changing conditions and also cannot provide alternative streams.

(Inventive) For the broadcast scenario the idea is that RTCP messages are generated in a network node, preferably the RNC for WCDMA (BSC for GPRS?). This is basically the logical location for generating the RTCP reports. The new RTCP receiver report is created based on radio related information. Therefore, the function of creating and sending an RTCP receiver report and the data collection function can be split.

Different RTCP reports do exist. In particular these are
Sender report (SR)
Receiver report (RR)
Source Description Items (SDES)
Bye-Message (BYE)
Application specific functions (APP)

Messages can be bundled to form a so-called compound message. Each compound message must contain a SDES RTCP message.

For the invention in particular the Receiver report (RR) is important. This is the message which needs to be received by the sender in order to adapt to changing bandwidth conditions.

We propose that such receiver reports are generated in e.g. the RNC based on the knowledge the RNC has about the link condition in one or more cells. The RNC could generate on message for each cell it is responsible for or even form a single message for all cells.

The RR messages include the following fields[1]:

SSRC_n (source identifier): 32 bits
The SSRC identifier of the source to which the information in this reception report block pertains.

fraction lost: 8 bits
The fraction of RTP data packets from source SSRC_n lost since the previous SR or RR packet was sent, expressed as a fixed point number.

cumulative number of packets lost: 24 bits
The total number of RTP data packets from source SSRC_n that have been lost since the beginning of reception.

extended highest sequence number received: 32 bits
The low 16 bits contain the highest sequence number received in an RTP data packet from source SSRC_n, and the most significant 16 bits extend that sequence number with the corresponding count of sequence number cycles.

interarrival jitter: 32 bits
An estimate of the statistical variance of the RTP data packet interarrival time, measured in timestamp units and expressed as an unsigned integer. The interarrival jitter J is defined to be the mean deviation (smoothed absolute value) of the difference D in packet spacing at the receiver compared to the sender for a pair of packets.

last SR timestamp (LSR): 32 bits
The middle 32 bits out of 64 in the NTP timestamp (as explained in Section 4) received as part of the most recent RTCP sender report (SR) packet from source SSRC_n. If no SR has been received yet, the field is set to zero.

delay since last SR (DLSR): 32 bits
The delay, expressed in units of 1/65536 seconds, between receiving the last SR packet from source SSRC_n and sending this reception report block. If no SR packet has been received yet from SSRC_n, the DLSR field is set to zero.

The values for the entries in the RR need to be generated. The first one is simply the sender ID, which is known.

(Inventive) For the second one, the fraction of lost packets, there are different alternatives. An appropriate value could be either the loss fraction seen by the RNC, or an estimate by the RNC depending on the current cell situation (e.g. radio resource usage, interference, …) and on the reliability level chosen for the transmission in the cell.
Third, the cumulative number of packet losses needs to be chosen according to the concept used for the previous one to avoid a mismatch. For example, if the loss fraction is based on the packet losses seen by the RNC, they should be used for this entry as well.
The highest sequence number received should be the highest one the RNC has seen.
The interarrival jitter could be either based on the jitter observed by the RNC, or probably better by an estimated value taking the link parameters into account (e.g. whether ARQ or repetition coding or FEC are used to ensure a certain degree of reliability).

4.2.2 Multicast Scenario

The multicast scenario differs from the broadcast scenario mainly in that a return channel is available. This as such would make end-to-end RTCP signaling possible, but because of problems indicated in chapter 0, it is here proposed to generate the RTCP messages at a network node, preferably the RNC for WCDMA (BSC for GPRS?). There are then two possibilities with the current version of RFC1889:

1. The UE generated RTCP messages are discarded in the UE and generated "from scratch" in the RNC (logically) or an intermediate node. With the new revision of RFC1889, the source can indicate to the clients, that no RTCP receiver reporting shall be used.

2. The UE generated RTCP messages are transmitted over the radio interface to the RNC (logically), but modified in the RNC according to certain principles described below. The RTCP message interval for RTCP messages from the UE can be larger than the RTCP message interval for RTCP messages from the intermediate node to the sender. The UE may even send RTCP messages only event driven, e.g. when certain values a out of range.

The input for setting the different fields of the RR messages are the same as for the broadcast scenario. To overcome the problems described in chapter 0, the following principles can be followed when setting the fields in the RR messages:

- When using a common transport channel in WCDMA (which is at the moment the assumption for both multicast and broadcast), there will always be some receivers with poor channel conditions suffering from large packet loss, while others get good quality. The RNC could in

this case 'shield' the poor receiver reports to maintain quality for the good users.

- If RNC detects an overload in a cell and wants to reduce the bit rate on the common channel used for MBMS, it can use the RTCP reports to signal this to the Multicast server. This can be either by increasing the measured packet loss ratio in the reports, or just reducing the highest sequence number received. This will be quicker than end-to-end signaling because of the radio interface latency and since per user RTCP reporting becomes rare for large user groups, as described in chapter 2.3.

4.2.3 General Handling

4.2.3.1 Logical Unit for RTCP Reports Generation

In the above description the RTCP report generation is done in the RNC. In general, this report generation and all related functionality is a logical function that could also reside in other network entities, such as e.g. the MB-SC. Dedicated protocols could be used to forward the relevant information from the RNC to the MB-SC in that case.

4.2.3.2 Achieve User Anonymity

Different from some Multicast applications in the fixed Internet like audio and video-conferences, mobile users form typically an anonymous community. Users in mobile networks which look at the same video clip are most likely not interested in knowing the names of other viewers and might also not be interested in revealing their identity.

RTCP messages (RR and SDES messages) according to the standard should (which is a clear recommendation to do so)

include the identity of users in e.g. the format of his/her email-address.

As discussed above, this is not desirable for the wireless multicast scenario, where users would like to maintain their anonymity and where they also do not want to pay for received information they are not interested in (irrelevant information).

Thus, the invention includes to transmit the RTCP messages, which are generated either in UEs or the intermediate node, only back to the RTP stream sender.

4.2.3.3 Number of Destinations Covered by Aggregated RTCP Report

Together with the generation of an aggregated RTCP report, the number of destinations for which the RTCP report applies should optionally be added to the report. This information is then potentially considered by the source in case of multicast stream adaptations. E.g. if an aggregated RTCP report covers thousands of destinations, the source could adapt for these destinations. If it covers only 10 destinations in a session where thousands of destinations are involved (e.g. in other cells), it may be better to advise the clients to switch to a unicast session instead.

4.2.4 RTCP handling in the Intermediate Node

Each compound RTCP packet must include a Report packet (SR or RR) and an SDES CNAME packet. To distinguish between several multicast sources, each receiver report (RR) packet is addressed by the SSRC of the source. Therefore, the Intermediate node must process and forward the downstream-multicast traffic (from the source to the receivers) and extract the SSRC of the Multicast Source. The SSRC is necessary to address the upstream RTCP receiver report packets, which are generated in the intermediate node. The

number of mobile terminals per cell (plus reception conditions per terminal) is provided by the RNC.

The Intermediate Node must allocate an SSRC identifier for each mobile terminal. The intermediate node must allocate the SSRC on behalf of each mobile terminal like described in [RFC1889] chapter 8. Beside the SSRC, the intermediate node must also provide a SDES CNAME item for each MT. There are several options of choosing the CNAME for a particular user:

- In case of anonymous participation (Source shall not get a clear CNAME), the SDES CNAME item is randomly chosen. Is can be in form of <random-number>@host, similar to the form described in 6.5.1 of [RFC1889]. The CNAME must be unique.
- In case of a non-anonymous CNAME, either the operator predetermines the user name (e.g. phonenumber@domain) or the user specifies the CNAME in a preference database.

Therefore, the intermediate node must maintain a list of MTs per cell plus the associated SSRCs (allocated by the intermediate node) and CNAMEs. The intermediate node functionality is possibly included in the BM-SC or the GGSN. The content of each RTCP packet is created like described in the previous chapters.

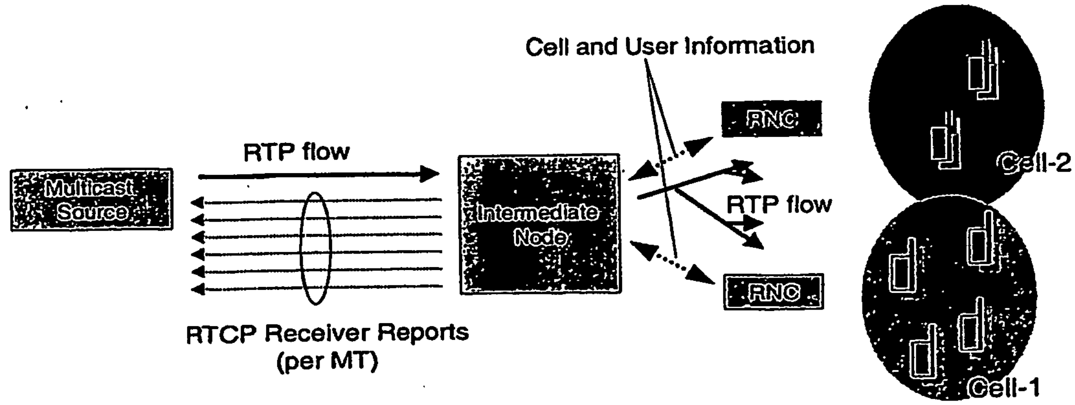


Figure 2: RTCP receiver report per Mobile Terminal

In case the very large user group, which is spread over a very large number of cells, the Intermediate node shall send RTCP packets not per MT but per cell. It is very likely, that each cell servers an approximately equal number of group members.
In this case, the Intermediate node allocates valid SSRCs and CNAME for each cell, which contains group members. The number of send RTCP packets is decreased. The transmission interval of RTCP packets and therefore also the reaction time of the multicast source decreases by sending per-cell RTCP packets.

As an additional RTCP packet type can be introduced into to weight the RTCP receiver report to the number of users in the cell. This RTCP packet type must part of a RTCP compound packet whenever this RTCP packets contains receiver reports for more than one member.

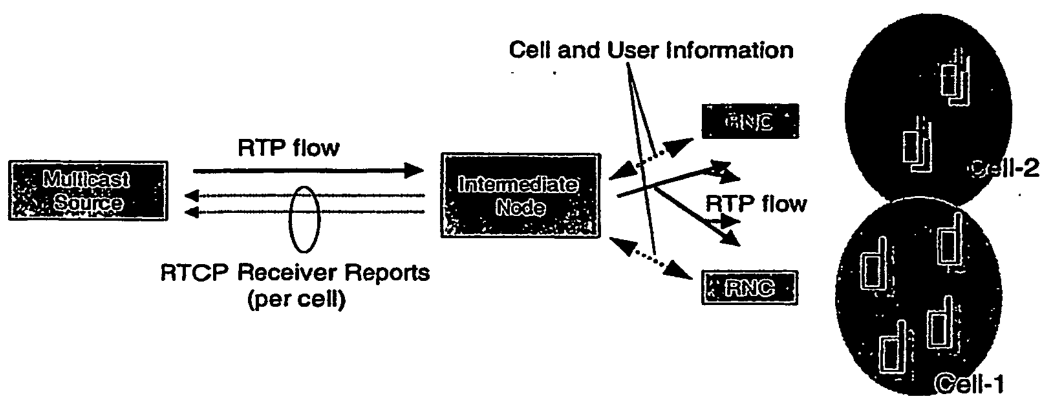


Figure 3: RTCP receiver report per cell

5. BENEFITS OF THE INVENTION

This invention solves all the problems as mentioned in chapter 3. In particular, the invention has the following benefits:

- o Aggregated RTCP reports, i.e. one RTCP report for all clients served in the same cell or all clients served by the same RNC.
- The RTCP reports optionally contain counters with the number of clients that are aggregated in the reports
- This is the only way of reporting in case no uplink channel is available, as in the simple broadcast solution that Ericsson is targeting in 3GPP R6.
- This will reduce the load on the air-interface and in the backbone network. However, it will very much reduce the load and the complexity in the source (e.g. hundreds or thousands of users in the same cell)
- The aggregated RTCP report can consider all information in the cell, rather than a single dedicated RTCP report per client. E.g. in case of 'almost' overload situations, the RNC knows this (before congestion is encountered by loosing frames) and can report this to the source for all clients in the same cell. This would improve the quality experience for the end-user.
- In general the reporting is faster since the air-interface delay is skipped. This is very much the same as for the single-user streaming case.
- Due to that the reports are aggregated and apply to multiple clients (often even all clients, e.g. in case of geographical area related multicast sessions) the source now has a means to adapt

to the clients, which was not the case without the aggregation.

- o Wireless and multicast specific RTCP reporting
  - Wireless and multicast specific RTCP reports (info to be added to the IETF RFCs and drafts) will take the specifics of wireless networks into account and will improve the overall service quality.
  - Other information from lower layer protocols as already known in the RNC, can be considered in the reports to further optimize the service quality. However, it is just one part of the invention to forward additional radio related info to the source. Another one is that the RNC considers the info (because that's where it has the knowledge) and makes an adaptation proposal (see above) to the source. With this the source doesn't need to have info about the radio network, since it's the radio network that converts this info to a known adaptation proposal for the source.

6.TERMINOLOGY AND ABBREVIATIONS

IETF	Internet Engineering Task Force
MBMS	Multimedia Broadcast/Multicast Service
MB-SC	Multicast Broadcast Service Center
QoS	Quality of Service
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
UDP	User Datagram Protocol
UE	User Equipment
WG	Working Group

7. ENCLOSURES

Ref[1] Schulzrinne, H., Casner, S., Frederick, R., Jacobson, V., 'RTP: A Transport Protocol for Real-Time Applications', Request For Comments 1889, Internet Engineering Task Force, 1996.

Ref[2] TS 22.146 Multimedia Broadcast / Multicast Service, Stage 1 (Release 5), version 5.2.0 (2002-03) ftp://ftp.3gpp.org/specs/latest/Rel-5/22_series/22146-520.zip

TS 23.846 Multimedia Broadcast / Multicast Service, Architecture and Functional Description (Release 5), version 0.3.0 (2002-01) ftp://ftp.3gpp.org/specs/Latest-drafts/23846-051.zip

Ref[3] TS 26.233 Transparent end-to-end packet switched streaming service (PSS); General description Release 5, version 5.0.0 http://www.3gpp.org/ftp/Specs/latest/Rel5/26_series/26233-500.zip

Ref[4] TS 26.234 Transparent end-to-end packet switched streaming service (PSS); Protocols and codecs Release 5 version 5.0.0 http://www.3gpp.org/ftp/Specs/latest/Rel-5/26_series/26234-500.zip

Ref[5] Schulzrinne, H., Casner, S., Frederick, R., Jacobson, V., 'RTP: A Transport Protocol for Real-Time Applications', Internet Draft, draft-ietf-avt-rtp-new-11.txt, November 2001.

Claims

1. Method for reporting on multicast/broadcast transmission in network wherein a multicast source provides the multicast/broadcast transmission to users being registered to multicast groups

characterised in that

- group structures of the multicast groups are determined and,
- real time determination of distribution characteristics is performed considering cell related characteristics determination and the group structure and,
- one feedback report per multicast group based on the real time determination of distribution characteristics is generated wherein said feedback report includes additional information including the number of users to which said feedback report applies and group-related adaptation proposal, considering the characteristics of the access network and,
- the multicast source utilizes the group feedback report by considering the percentage of the users for which said feedback report applies and,
- the multicast/broadcast transmission is adapted accordingly to the utilized feedback report.

2. Method according to claim 1 characterised in that the determination of the group structures considers speed of the multicast/broadcast transmission.

3. Method according to claim 1 characterised in that a negotiation on the applied feedback reporting mechanism

between the network and the users is performed at the beginning of a session.

4. Method according to claim 1 characterised in that the group-related adaptation proposal, considering the characteristics of the access network proposes multiple unicast if that's possible handover overlay cell or other access network

5. Method according to one of the previous claim characterised in that the real-time determination of the distribution characteristics is performed on the level of a geographical area having one or several radio cells and is applicable to the group of users.

6. Method according to one of the previous claim characterised in that the group structure of the users includes all users in the same cell or all users in the same group of cells and with similar conditions like for example same client terminal type, same Quality of Service, same time of registration.

7. Method according to one of the previous claim characterised in that the additional information of the feedback report includes group-related adaptation proposal, considering the characteristics of the access network.

8. Method according to one of the previous claim characterised in that the adaptation of the multicast/broadcast transmission includes the announcement of a new transmission channel to the users or reducing of the bit-rate or switching to more reliable codecs.

9. Intermediate node adapted to perform reporting on multicast/broadcast transmission wherein a multicast source provides the multicast/broadcast transmission to users being registered to multicast groups

characterised by

-means for providing real time determination of distribution characteristics considering cell related characteristics determination and the group structure and,

-means for generation of feedback report per multicast group wherein said feedback report includes additional information including the number of users to which said feedback report applies and

-means for sending the feedback report to the multicast source

10. Multicast source adapted to perform reporting on multicast/broadcast transmission wherein a multicast source provides the multicast/broadcast transmission to users being registered to multicast groups

characterised by

- means for utilization of a received group feedback report by considering the percentage of the users for which said feedback report applies and,
- means for adaptation of the multicast/broadcast transmission accordingly to the utilized feedback report.

11. System adapted to perform reporting on multicast/broadcast transmission wherein a multicast

source provides the multicast/broadcast transmission to users being registered to multicast groups

characterised in that

said system has multicast source according to claim 9, intermediate node according to claim 8 means for determination of group structure and means for real time determination of distribution characteristics considering cell related characteristics determination and the group structure.

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Abstract

The invention relates to a method, intermediate node, multicast source, and system for reporting on multicast/broadcast transmission wherein a multicast source provides the multicast/broadcast transmission to users being registered to multicast groups. The basic idea is to have an intermediate node in the wireless network taking care of an 'aggregated' RTCP reporting, rather than that each user individual performs its own reports. The aggregated' RTCP reporting or general one feedback report per multicast group is generated. A real time determination of distribution characteristics is performed considering cell related characteristics determination and the group structure and. The generation of the feedback report is based on the real time determination of distribution characteristics wherein said feedback report includes additional information including the number of users to which said feedback report applies. Said feedback report is sent to the multicast source, which utilizes the group feedback report by considering the percentage of the users for which said feedback report applies. The multicast/broadcast transmission is adapted accordingly to the utilized feedback report.